propellerhead

EUROPA BY REASON

OPERATION MANUAL

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Europa by Reason

Installing Europa By Reason

The Europa by Reason synthesizer comes as a 64-bit VST2 plugin for Windows and macOS - and also as a 64-bit AU plugin for macOS.

Windows

1. **Unzip the downloaded zip file and extract the "EuropaSetup.exe" file to your Desktop.**
2. **Double click the "EuropaSetup.exe" file and follow the instructions in the Setup Wizard.**
 The default location for the installation will be in the C:\Program Files\VSTPlugins folder. If you want to install in another location, choose one before clicking the Next button. The default location of the "Europa Patches" content folder will be in: C:\Users\[your user]\Documents\Europa by Reason\. The "Europa Patches" folder contains patch category sub-folders and a "Samples" folder with User Wave samples.
 If you want to have the content folders in another location, choose one before clicking the Next button.
3. **Launch your DAW and add an Europa instrument, as described in "Adding Europa by Reason in your DAW".**

macOS

The macOS installer includes both the VST2 and AU versions of the plugin.

1. **Unpack the downloaded zip file.**
2. **Double click the Europa by Reason.pkg file and follow the instructions in the installer.**
 This will install the VST2 version in the default /Library/Audio/Plug-Ins/ folder and the AU version in the default /Library/Audio/Plug-ins/Components/ folder. Patches and samples will be installed in: /Library/Audio/Presets/Propellerhead Software/
3. **Launch your DAW and add an Europa instrument, as described in "Adding Europa by Reason in your DAW".**

Adding Europa by Reason in your DAW

The Europa by Reason instrument should be added as an Instrument/MIDI track in your DAW - just like when adding any other synthesizer VST/AU instruments. For more details on how to add VST/AU instruments, please refer to your DAW's User Manual/documentation.

Authorizing your computer to run Europa by Reason

After purchasing Europa by Reason, you will get a license number consisting of 8 x 8 characters. You can find it at any time under the VST Plugins tab on your Propellerhead account, at https://www.propellerheads.se/user/products/.

When you first add an Europa by Reason instance in your DAW, a dialog will tell you that no license was found:

- → **Click "Enter License", paste your license number into the dialog that appears, and then click OK.**
- **Alternatively, you can run Europa by Reason in Demo Mode.**
 The plugin will then time out after 20 minutes, and saving patches is disabled.

About uninstalling Europa by Reason

Should you want to uninstall Europa by Reason, proceed as follows:

- → **Under Windows, use the standard Uninstall function on the "Settings > Apps & features" page.**
- → **Under macOS, delete the "Europa.vst" file in /Library/Audio/Plug-Ins/ and the "Europa.component" file in /Library/Audio/Plug-ins/Components/. Also delete the folders "Europa Patches" and "Samples" in /Library/Audio/Presets/Propellerhead Software/.**

Introduction



The Europa Shapeshifting Synthesizer is the most advanced and sonically “wide” synthesizer in Reason. Despite being a very advanced synthesizer, it’s really easy to create great sounds from scratch. Just a few mouse clicks and knob twists in a Sound Engine section will generate truly impressive and inspiring sounds!

The three powerful and flexible sound engines offer a unique combination of analog/wavetable/spectral/physical modeling/FM synthesis techniques. If you like, you could also draw your own waveforms and filter curves to design your own completely unique sounds. In addition to this you can also load your own sample into Europa and use as a wavetable and/or filter! Each sound engine also has its own Unison module for generating really wide multi-voice stereo chorus effects.

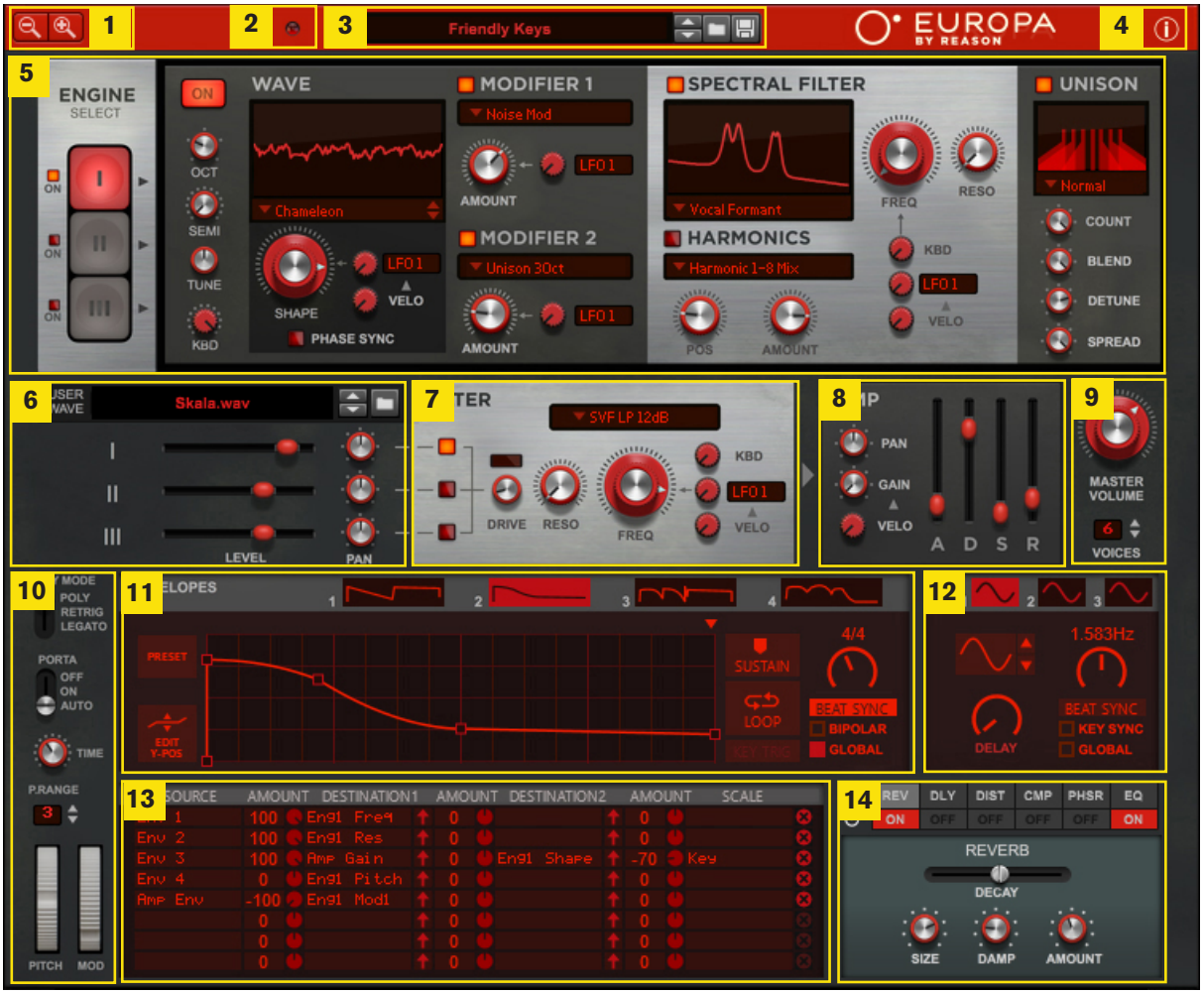
The extensive Envelopes section and Modulation Bus section allow for very detailed and flexible modulation and control. Europa also features a top-notch semi-modular multi-effect section so you could put that final touch on your sounds.

The VST2/AU version of the Europa Shapeshifting Synthesizer - Europa by Reason - lets you enjoy Europa also if you are using other DAWs than Reason!

Don’t forget to check out the Europa Shapeshifting Synthesizer video here!

Panel overview

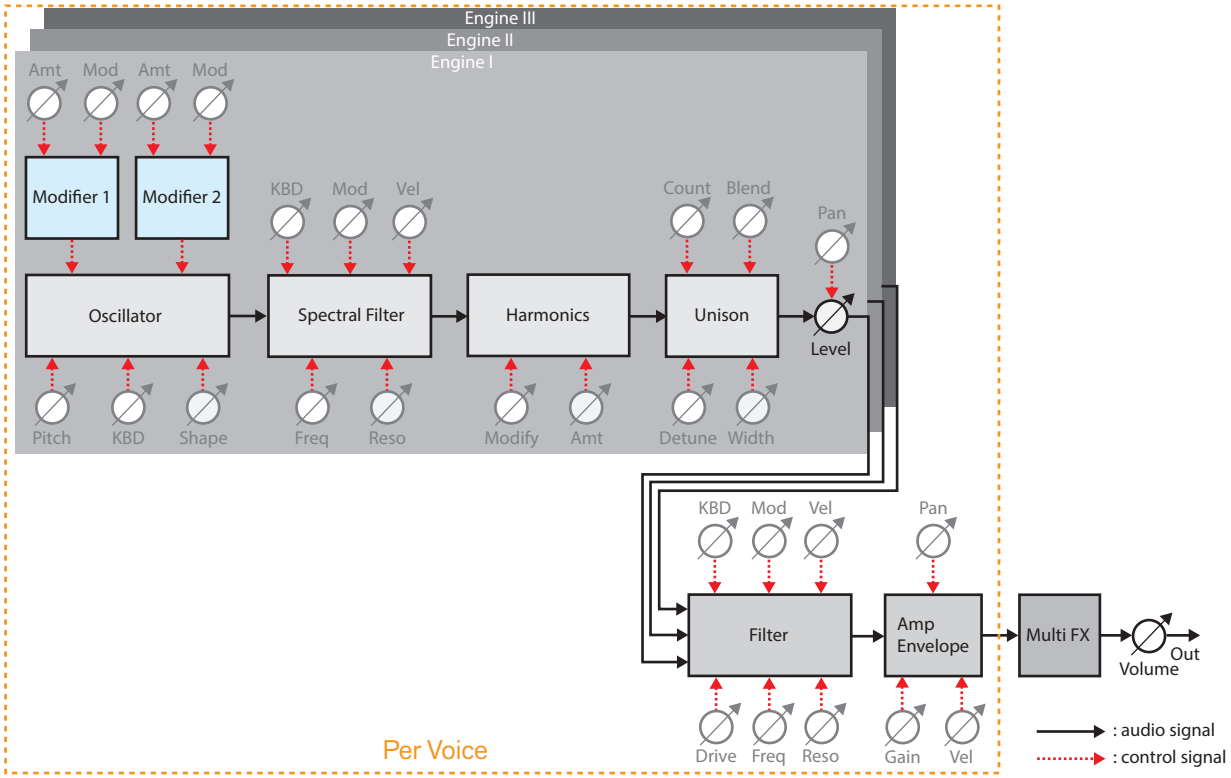
The Europa front panel contains the following sections:



- **1. Zoom In/Out buttons for changing the size of the panel.**
- **2. MIDI Note On LED.**
- **3. Patch Selector (for browsing, loading and saving patches).**
- **4. Info button. Click this to view information about Europa by Reason!**
- **5. Sound Engines section.**
- **6. User Wave and Mixer section.**
- **7. Filter section.**
- **8. Amplifier section.**
- **9. Global output controls.**
- **10. Global performance and "play" controls.**
- **11. Envelopes section.**
- **12. LFO section.**
- **13. Modulation Bus section.**
- **14. Effects section.**

Signal flow

The picture below shows the basic signal flow in Europa:



Europa signal flowchart.

- **The "hearts" of Europa are the three Sound Engines I, II and III.**
- **First in each Sound Engine is an Oscillator, which generates the basic audio signal.**
 The oscillators in Europa are extremely powerful and flexible. Besides all the basic "analog" waveforms, the oscillators can also generate a vast variety of wavetable waveforms, physical modeling signals and other types of unique signals - and also your own samples! The signals can also be continuously transformed into various shapes.
- **The Oscillator signal can be modified by the two Modifiers.**
 The Modifiers feature a huge amount of algorithms that can modify the Oscillator signal in various ways.
- **The signal from the Oscillator is routed to the Spectral Filter.**
 The Spectral Filter affects the partials of the signal. The algorithms could be various types of filters - or special purpose signal processors.
- **The signal from the Spectral Filter can then be routed to the Harmonics processor.**
 The Harmonics processor modulates the harmonics in the signal, for example introducing ensemble or stretch effects.
- **The signal from the Harmonics processor can then be routed to the Unison module.**
 This module can generate various types of unison chorus effects, to make the sound really fat and wide.
- **The signals from the Sound Engines are then routed to the Mixer, where you can set the mix between the three Sound Engine output signals and also pan the signals individually.**
- **The mixed signal is then routed to the Filter, Amp Envelope and then, via the Multi FX section, to the stereo outputs.**
- **The remaining sections in Europa (Envelopes 1-4 and LFOs 1-3) can be freely assigned to modulate destination parameters via the Modulation Bus section.**

Playing and using Europa

Loading and saving patches



Loading patches

1. **Click the "Browse patch" (folder icon) button.**
 This will open the default Europa Patches folder, which was installed during the Europa plugin installation.
2. **Double click a patch in the folder - or select a patch and click Open - to load the patch in Europa.**

→ **To load other patches from the last folder you could also click the Select previous/next patch buttons (Up/Down arrow buttons) on the Europa panel.**

→ **Alternatively, click on the patch display and select a patch from the list than appears.**
At the top left of the list is a patch called "Init Patch". Selecting this will initialize the Europa parameters and let you begin creating your own patch from scratch.

Saving patches

1. **Click the "Save patch" (disk icon) button.**
 This will open the default Europa Patch folder.
2. **If desired, rename your patch, select a destination folder and click Save to save your patch.**

Global output controls



Master Volume

This is the main stereo output volume control.

Voices

Here you set the desired maximum polyphony of your patch, from 1 to 16 voices. This control is mainly intended for deliberately restricting the polyphony of a sound. If you just want to play a patch polyphonically you can leave the Voices setting at 16 at all times. The CPU Load won't increase with higher voice number settings - only if you play a lot of notes simultaneously.

→ **If you want monophonic playback you could use the "Retrig" and "Legato" modes instead of lowering the Voices parameter to 1.**

Global performance and “play” controls



Key Mode

Here you choose how Europa should respond to MIDI Note data:

- **Poly**
 Select this if you want to play Europa polyphonically. The maximum number of voices is 16. The number of voices is set in the Voices control at the center right of the Europa panel, see “Voices”.
- **Retrig**
 Select this if you want to play Europa in monophonic mode and always retrigger the envelopes as soon as you play a new note.
- **Legato**
 The Mono Legato mode is also monophonic. However, if you play a new note without having released the previous one, the envelopes won’t start over.

Porta

Portamento makes note pitches glide from previous notes to new ones, at the time set with the Time knob. Portamento can be used in all Key modes (see above).

- **When On in Poly Key Mode (see above), the pitches will glide from any of the available voices.**
 The results will be unpredictable since there is no way of controlling from which note(s) the glide(s) will commence. The effect is very nice, though.
- **When On in Retrig or Legato Key Mode (see above), the pitch will glide between consecutive notes.**
- **In Auto mode, the pitch will glide between consecutive monophonic notes only when you play legato. If you have selected Poly Key Mode (see above), Auto will have no effect at all.**
 If you release the previous key before hitting the new key, there will be no portamento effect.

P.Range

→ **Set the desired Pitch Bend range for the “Pitch” wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-24 semitones (+/-2 octaves) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Europa also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the “P.Range” control above the Pitch bend wheel.

Mod

The Mod wheel can be used for controlling almost any parameter in Europa. Use the Mod wheel as a Source parameter in any of the Modulation Source boxes in the different sections. Or use it as Source parameter in the Modulation Bus section and then route to the desired Destination parameter(s), see "The Modulation Bus section".

Panel reference

Sound Engines On/Off and Edit Focus section



Engine Select

- **Click the On LED buttons to activate the corresponding Sound Engine.**
- **Click the I, II or III LED radio buttons to select the corresponding Sound Engine for editing.**

The Oscillator section



Here is where you choose oscillator waveform and set the wave shape and pitch for the selected Sound Engine.

On

- **Click the red rectangular LED button to switch the selected Sound Engine on/off.**

Oct

- **Set the pitch in octave steps.**
 Range: 5 octaves.

Semi

→ **Set the pitch in semitone steps.**
Range: 12 semitones (one octave).

Tune

→ **Change the pitch in steps of 1 cent.**
Range: +/- 50 cents (down or up half a semitone).

Kbd

→ **Set how much the pitch should track incoming MIDI Notes.**
Range: 0% (no tracking (constant pitch)) to 100% (1 semitone per note).

Waveform display

The interactive Waveform display shows the waveform shape in real-time.

- **Clicking and dragging vertically in the display changes the Shape parameter, see "Shape".**
- **Clicking and dragging horizontally in the display changes the Modifier 1 Amount parameter, see "Amount".**

Waveform selector

→ **Click the Waveform name box to bring up a menu of the available waveforms.**
The wave shapes are shown in the display above and are updated in real-time according to the current settings and modulations. A great way to understand how the sound actually "looks".

The waveforms are:

- **Basic Analog**
A pure sinewave at Shape=0%, gradually transformed via triangle and square towards a sawtooth wave at Shape=100%.
- **Square-Ramp**
A square wave at Shape=0%, gradually transformed towards a sawtooth wave at Shape=100%.
- **Saw-Triangle**
A negative ramp sawtooth wave at Shape=0%, gradually transformed via triangle towards a positive ramp sawtooth wave at Shape=100%.
- **Pulse Width**
A 0% duty cycle pulse wave (silence) at Shape=0 gradually transformed via a 50% duty cycle square wave towards a 100% duty cycle pule wave (silence) at Shape=100%.

► **Modulate the Shape parameter from an LFO to achieve PWM, see "Shape Modulation" below.**

- **Game**
A lo-fi "early computer game" type of signal. Turn the Shape knob to change the overtone contents and the octave transposition.
- **Synced Sine**
A pure sinewave at Shape=0%. As the Shape is increased, the pitch of the synced sinewave oscillator is raised.
- **Formant Sweep**
A cosine window modulated by a sinewave. Turn the Shape knob to change the sinewave frequency and thus sweep through the generated formants.
- **Electro Mechanical**
This is a simulation of an electric piano. A soft/mellow tone at Shape=0% gradually transformed towards an agitated signal at Shape=100%, with natural sound at the 12 o'clock position (50%).

- **Vocal Cord**
 A simulation of a vocal cord with a bit of noise modulation. Change the overtone content with the Shape knob.

► **Try this together with the Vocal Formants algorithm in the Spectral Filter section to generate "vocal" sounds, see "Vocal Formant".**

- **Karplus-Strong**
 A physical model of a "string", generated by sending a short pulse through pitched delay lines. At Shape=0% there is no damping and at Shape=100% there is full damping, which results in just a short clicking sound.

► **Try this together with the "Stretch" algorithm in the Harmonics section to create realistic metallic sounds.**

- **Envelope 3-4**
 This is a special mode where you can manually draw your waveforms in the Envelope 3 and Envelope 4 windows and then gradually crossfade between the drawn waveforms using the Shape knob. See "Using the Envelope 3 and Envelope 4 curves as Sound Engine waveforms" for information on how to draw your own waveforms.
- **FM > FM Ratio (1:1, 1:2, 1:8, 2:1)**
 These are frequency modulated sine waves with different frequency ratios between the carrier (C:) and modulator (:M) signals. Set the frequency modulation amount with the Shape knob.
- **FM > FM Feedback**
 A pure sinewave signal at Shape=0% gradually fed back internally at an 1:1 ratio. The feedback signal is filtered before fed back to the carrier signal. If you modulate the Shape parameter from e.g. an LFO you will get a similar result as when using the FM FB Noise waveform without Shape modulation, see "Noise > FM FB Noise" below.
- **Noise > S/H Noise**
 A sample & hold modulated noise. Change the sample & hold rate with the Shape knob. If you play high up on the keyboard at high Shape values, you get a kind of "pitched noise" sound.

► **To get white noise, set Shape to max, set Oct to -1 and turn Kbd to 0.**

- **Noise > Perlin Noise**
 A pure sinewave signal modulated by low frequency noise. At Shape=0% the noise has its lowest frequency and at Shape=100% the noise frequency is higher (but still low-frequent). The character of the signal is similar to the Band Noise in the Thor synthesizer.
- **Noise > Bit Noise**
 This generates a random lo-fi "digital" bit noise. At Shape=0% the signal is completely silent and with increasing Shape values the signal is modulated faster and in a wider frequency range.
- **Noise > FM FB Noise**
 A pure sinewave signal at Shape=0% gradually fed back internally at an 1:1 ratio. The feedback signal unfiltered before fed back to the carrier signal which gives the signal a noisy character.

► **To get a cleaner FM signal, use the FM Feedback waveform, see "FM > FM Feedback" above.**

- **Noise > Freeze Noise**
 This signal produces a range of noises, from tonal noise up to almost white noise, by amplitude modulating the signal's partials with noise.
- **Wave Tables >**
 The Wave Tables sub-menu contains a selection of very useful wave tables. Each wave table features eight waveforms that you could crossfade between with the Shape knob.
- **User Wave/User Wave Smooth**
 The User Wave options let you use the external sample you have loaded in the User Wave section (see "The User Wave and Mixer section"). The oscillator then generates and plays back wavetables (grains) of that sample. The "User Wave Smooth" algorithm uses a crossfaded loop within each grain, which produces a smoother character to the sound. Set the playback position in the sample with the Shape knob. Modulate the Shape parameter, for example from a negative Envelope ramp, for continuous movement in the sample.

Shape

→ **Turn the Shape knob to change the shape of the currently selected waveform.**
The wave shapes are shown in the display above and are updated in real-time according to the current Shape settings.

Shape Modulation

→ **Click the Shape Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

→ **Set the modulation amount with the Shape Modulation Amount knob.**

→ **Turn the Velo knob to control the Shape Modulation Amount from Keyboard Velocity.**

→ **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**

Phase Sync

→ **Click the Phase Sync button to force the waveform cycle to always start at the same phase (0 degrees).**
When active, the sound character will be the same each time you play the same note. When inactive, the sound character will vary more or less each time you play the same note.

The Modifiers section



The two Modifiers can be used for modifying the currently selected waveform in various ways. The two Modifiers are identical in functionality and can be used alone or together (or not at all).

Modifier On/Off

→ **Click the On/Off LED buttons to activate/deactivate the corresponding Modifier.**

Modifier selector

→ **Click the Modifier name box to bring up a menu of the available Modifier types.**
The available Modifier types are:

- **Faded Sync**
 This is oscillator sync but with a crossfade at the sync positions. This makes the effect a little smoother (less bright) than with regular hard sync, see below.
- **Hard Sync**
 Oscillator hard sync is when one oscillator restarts the period of another oscillator, so that they will have the same base frequency. If you change or modulate the frequency of the synced oscillator you get the characteristic sound associated with oscillator sync. Control the frequency of the synced oscillator (and thereby the overtone spectrum) with the Amount knob.

- **Invert**
 This inverts the waveform phase at a variable position within the waveform cycle. Set the phase angle with the Amount knob.
- **Mirror**
 This mirrors the waveform cycle (in the time line) at a variable position in the waveform cycle. Set the mirroring position in the waveform cycle with the Amount knob. At Amount=50% the waveform is completely symmetric.
- **DownSample**
 This lets you quantize the waveform in time, i.e. reduce the sample rate. Set the sample rate reduction amount with the Amount knob.
- **Quantize**
 This lets you truncate the signal's bit depth, thus making it possible to achieve that noisy, characteristic "8-bit sound" for example. Set the bit-reduction amount with the Amount knob.
- **Phase Distort**
 This distorts the waveform by modulating the start phase of the waveform cycle. This generally creates a brighter tone towards the extremes of the Amount range (0% and 100%). At Amount=50% the signal is unaffected.
- **Self Multiply**
 This multiplies a copy of the waveform with the original waveform. Set the phase angle of the copied waveform with the Amount knob.
- **Noise Mod**
 This modulates the waveform with low frequency noise. Perfect for adding e.g. "breath noise" to a signal. Set the noise modulation amount with the Amount knob.
- **Shaping > Wrap**
 This amplifies the signal above the available headroom and then wraps the peaks down into the available headroom. This adds quite an aggressive distortion to the sound.
- **Shaping > Fold**
 This amplifies the signal above the available headroom and then "mirrors" the peaks down into the available headroom. Fold is similar to the Wrap shaping but is generally less "aggressive".
- **Shaping > Hard Clip**
 This amplifies the signal above the available headroom and then clips the peaks that are above the headroom. Generally, a signal that is clipped to the maximum would result in a pulse/square shaped waveform.
- **Shaping > Soft Clip**
 Soft clip is similar to hard clip described above, but has a smoother shape at the clipping points and thus generates less overtones.
- **Shaping > Sine Shape**
 This generates sine shaping distortion to the signal.
- **Shaping > Glitch 1**
 This distorts the waveform by introducing a short low-frequency noise glitch in the waveform cycle, but only in parts of the waveform that go from zero to positive level.
- **Shaping > Glitch 2**
 This is similar to Glitch 1 described above, but introduces a more high-frequent noise glitch in the waveform cycle.
- **Harmonics > Octave**
 This makes it possible to gradually crossfade between the original signal and a copy of the signal one octave above. Set the crossfade amount with the Amount knob.
- **Harmonics > Fifth**
 This makes it possible to gradually crossfade between "one octave up" and "one octave+one fifth up". As soon as you turn on the Fifth modifier you automatically raise the pitch by one octave. The reason for this is that this is done by multiplying frequencies, i.e. you crossfade between the double and triple of the original frequency. Set the crossfade amount with the Amount knob.

- **Harmonics > 16 Harmonics**
 This makes it possible to gradually crossfade between the original signal and copies of the signal at the 16 first harmonics above the original frequency. Set the position in the harmonic spectrum with the Amount knob.
- **Harmonics > Fund. + 16 Harm.**
 This is the same as "16 Harmonics" described above, except this always keeps the original signal mixed in with the selected harmonic.
- **Harmonics > All 16 Harm.**
 This gradually adds copies of the original signal at the first 16 harmonics. Turning up the Amount knob will add on the harmonics one by one until all 16 harmonics are present in the signal.
- **Harmonics > Ring Harm.**
 This multiplies the waveform with a sinewave signal to generate a ring modulator effect. Set the modulator frequency with the Amount knob.
- **FM > FM Ratio (1:1, 1:2, 1:8, 2:1)**
 These modifiers let you frequency modulate the currently selected Waveform at various ratios. The carrier signal is the currently selected Waveform (C:) and the modulator (:M) is the modifier signal. Set the frequency modulation amount with the Amount knob.
- **FM > FM Feedback**
 Here, an internally fed back sinewave signal at an 1:1 ratio modulates the waveform (same signal type as in the "FM > FM Feedback" Waveform).
- **Detuning > Unison3**
 This simulates 2 copies of the original signal. The Amount knob controls the detuning amount and rate.
- **Detuning > Unison7**
 This simulates 6 copies of the original signal. The Amount knob controls the detuning amount and rate.
- **Detuning > Ensemble**
 This simulates a variable number of copies of the original signal. The Amount knob controls the number of copies, the detuning amount and rate.
- **Detuning > Unison 3Oct**
 This simulates 2 copies of the original signal at +1 and +2 octaves relative to the original signal. The Amount knob controls the detuning amount and rate.
- **Formant**
 This simulates a formant (body) filter, which produces multiple peaks and notches in the frequency spectrum of the signal. The Amount knob controls the formant transposition in the frequency spectrum. At Amount=50% the signal is unaffected. Below 50% the formant is transposed down and above 50% it's transposed up.

 To make the formant static in the frequency spectrum, regardless of which note you play, modulate the Amount parameter using the "-KEY" (inverted) modulation source with a fairly high value (see "Amount Modulation" below). This is especially useful if you are using an acoustic instrument sample as User Wave in the Oscillator section.

Amount

→ **Turn the Amount knob to change the modification amount of the currently selected Modifier.**
The wave shapes are updated in real-time and shown in the Waveform display.

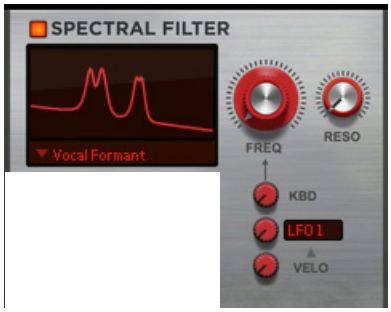
Amount Modulation

→ **Click the Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

→ **Set the modulation amount with the Amount Modulation knob.**

← **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**

The Spectral Filter



The signal from the Oscillator section can then be processed by the Spectral Filter. The Spectral Filter features a wide variety of algorithms that affect the partials of the signal.

Spectral Filter On/Off

→ **Click the On/Off LED button to activate/deactivate the Spectral Filter.**

Spectral Filter display

The interactive Spectral Filter display shows the filter shape in real-time.

- **Clicking and dragging vertically in the display changes the Freq parameter, see "Freq".**
- **Clicking and dragging horizontally in the display changes the Resonance parameter, see "Reso".**

Spectral Filter selector

→ **Click the Spectral Filter name box to bring up a menu of the available filter types.**
The available filter types are:

- **LP 12**
 This simulates a standard 12dB/octave lowpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.
- **LP 24**
 This simulates a standard 24dB/octave lowpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.
- **HP 24**
 This simulates a standard 24dB/octave highpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.
- **BP 12**
 This simulates a standard 12dB/octave bandpass filter. Set the center frequency with the Freq knob and the resonance amount with the Reso knob.
- **Par EQ**
 This simulates a standard single band parametric equalizer with a fixed bandwidth. Set the center frequency with the Freq knob and the gain/attenuation with the Reso knob.
- **Dual Peak**
 This simulates two 12dB/octave bandpass filter routed in parallel. Set the center frequency of the first bandpass filter with the Freq knob and the peak separation with the Reso knob.
- **Vocal Formant**
 This simulates the formants of the vocal tract by using multi-peak+notch filters. Change the formant with the Freq and Reso knobs.

- **LP Variable Slope**
 This simulates a non-resonant lowpass filter with a variable attenuation slope. Set the cutoff frequency with the Freq knob and the attenuation slope with the Reso knob.
- **HP Variable Slope**
 This simulates a non-resonant highpass filter with a variable attenuation slope. Set the cutoff frequency with the Freq knob and the attenuation slope with the Reso knob.
- **Comb +**
 This simulates a multi notch filter, great for phaser types of effects. Set the cutoff frequency of the first notch with the Freq knob and the attenuation amount - and consequently the bandwidth - of the notches with the Reso knob. The difference between "Comb +" and "Comb –" (see below) is in the position of the peaks in the spectrum. The main audible difference is that the "Comb –" version causes a bass cut.
- **Comb -**
 This simulates a comb filter with a positive feedback loop - but without feed forward - ideal for flanger and phaser types of effects. Set the cutoff frequency of the second peak with the Freq knob and the resonance amount with the Reso knob. The difference between "Comb +" (see above) and "Comb –" is in the position of the peaks in the spectrum. The main audible difference is that the "Comb –" version causes a bass cut.
- **Resonator 1,2 and 3**
 The three Resonator algorithms contain formant filter tables that simulate various body resonances (multi-peak+ notch filters). Set the position in the formant tables with the Freq knob and the resonance with the Reso knob.
- **Envelope 4**
 This is a special mode where you can manually draw your own filter curve in the Envelope 4 window. You then control the cutoff/center frequency with the Freq knob and the resonance with the Reso knob. See "Using the Envelope 4 curve as a Spectral Filter curve" for information on how to draw your own filter curves.
- **User Wave**
 This utilizes a filter generated from FFT analyses of the external sample you have loaded in the User Wave section (see "The User Wave and Mixer section"). Transpose the formant up/down in the frequency spectrum with the Freq knob and change the filter's position in the sample with the Reso knob.

► **To create a classic "vocoder" effect, load a vocal/speech sample in the User Wave section. Then, set the Freq knob to 50%, the Reso knob to 0% and the KBD knob to 0%. Then, have one of the Envelopes modulate the Reso parameter using the Modulation Bus (see "The Modulation Bus section"). Create a positive ramp (inverted sawtooth) envelope and set the Reso modulation amount to 100%.**

Freq

→ **Set the cutoff/center frequency of the currently selected Spectral Filter type.**

Reso

→ **Set the resonance amount of the currently selected Spectral Filter type.**

Frequency Modulation

→ **Turn the Kbd knob to set the keyboard tracking amount.**
At 0% the filter is static and doesn't track the keyboard at all. At 100% the filter tracks the keyboard 1 semitone per note. At values above 0% you can also see the filter curve move sideways in the Spectral Filter Display depending on where on the keyboard you play.

→ **Click the Frequency Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

→ **Set the modulation amount with the Frequency Modulation Amount knob.**

→ **Turn the Velo knob to control the Frequency Modulation Amount from Keyboard Velocity.**

► **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**

The Harmonics section



The Harmonics section offers extensive modulation possibilities of the partials of the signal. For most algorithms the partials' characteristics is displayed in the Spectral Filter display, see "Spectral Filter display".

Harmonics On/Off

→ **Click the On/Off LED buttons to activate/deactivate the Harmonics section.**

Harmonics selector

→ **Click the Harmonics name box to bring up a menu of the available harmonic algorithms.**
The available Harmonics types are:

- **Random Gain**
 This alters the gain for each of the partials in the signal in a random fashion. Turn the Pos knob to change the randomization "pattern" and the Amount knob to change the partial gain levels in the "pattern".
- **Harmonic 1-8 Mix**
 This alters the gain/attenuation for the first eight partials in the signal. Turn the Pos knob to crossfade between the partials and the Amount knob to change the partial gain/attenuation level. Amount levels below 50% attenuate all partials but the one selected with the Pos knob. Amount levels above 50% attenuate the partial selected with the Pos knob.
- **Odd-Even**
 This alters the gain/attenuation of the partials in the signal. At Pos=0% the Amount knob controls the mix strictly between the odd and even partials in the signal. At other Pos values, the gain/attenuation is not strictly on odd and even partials. At Amount=50% the Pos value has no effect.
- **Stretch**
 This stretches or squeezes all partials (overtones) - except for the fundamental - in the signal, up or down in the frequency range. Perfect for turning a harmonic signal into a more inharmonic one. Change the stretch amount with the Amount knob. The Pos knob controls the start phase of all the overtones. At Pos=0% all partials start at the same phase. When Amount is set fairly high the Pos parameter have little or no influence on the sound.

← **Try this with the Karplus-Strong waveform to create really realistic metallic sounds, see "Karplus-Strong"!**

- **Ensemble**
 This is the perfect algorithm for really dense pad sounds. The Ensemble algorithm simulates a type of chorus effect by utilizing noise modulation of the partials. Set the noise frequency with the Pos knob and the mix level with the Amount knob.
- **Ensemble Sparse**
 The Ensemble Sparse algorithm also utilizes noise modulation of the partials, but here a lot of noise frequency bands are cut out. This makes Ensemble Sparse sound more animated and less smooth than the Ensemble algorithm described above. Set the noise frequency with the Pos knob and the mix level with the Amount knob.
- **HF Noise**
 This amplitude modulates the high frequencies in the signal with (high-frequency) noise, perfect for adding "breath noise" to the signal, for example. Set the noise frequency with the Pos knob and the noise mix level with the Amount knob.

- **Harmonic Lag A-R**
 The Harmonic Lag A-R algorithm is designed especially for use with the User Wave algorithm in the Spectral Filter (see "User Wave") to create vocoder effects. The Harmonic Lag A-R algorithm controls the Spectral Filter - so the Spectral Filter has to be on for this to work!

 Note that the Harmonic Lag A-R algorithm works on the filter partials - not the oscillator's signal partials. Set the Attack time of the filter partials with the Pos knob and the Release time with the Amount knob. These controls work similarly to the Attack and Decay parameters on the BV512 Vocoder device.

Pos

→ **Turn the Pos knob to change the frequency of the currently selected Harmonics algorithm.**
The frequency spectrum is updated in real-time and shown in the Spectral Filter display.

Amount

→ **Turn the Amount knob to change the intensity of the currently selected Harmonics algorithm.**
The frequency spectrum is updated in real-time and shown in the Spectral Filter display.

The Unison section



The Unison function generates detuned duplicates of the signal in pairs on either side of the original signal's pitch.

Unison On/Off

→ **Click the On/Off LED button to activate/deactivate the Unison section.**

Unison display

The Unison display shows the unison characteristics, as set with the controls in the Unison section. Note, though, that this display is not interactive like the Waveform and Spectral Filter displays.

Unison Type selector

→ **Click the Unison name box to bring up a menu of the available Unison types.**
The available Unison types are:

- **Normal**
 This generates duplicates of the signal on either side of the original signal's pitch.
- **Fourth**
 This generates duplicates of the signal on either side of the fourth above the original signal's pitch.
- **Fifth**
 This generates duplicates of the signal on either side of the fifth above the original signal's pitch.

- **Octave Down**
 This generates duplicates of the signal on either side of the original signal's pitch - one octave down.
- **Phase Only**
 This generates duplicates of the signal on either side of the original signal's pitch. The Detune parameter (see below) now controls the phases of the signal copies, instead of the detuning. This is great for creating wide stereo sounds without lots of detuning.

 If Phase Sync is active in the Oscillator (see "Phase Sync"), all signals in the Unison function get the same relative start phases each time you play the same note. The original signal always has the phase 0 degrees.

Count

→ **Set the number of desired signals in the unison effect.**
Range: 1-7.
Note that for even numbers, the original signal is represented by two duplicates.

Blend

→ **Set the mix between the original signal and the duplicates.**
For even "Count" numbers (see above), one of the duplicates represents the original signal in the mix.

Detune

→ **Set the pitch detuning of the signal duplicates.**
If the "Phase Only" Unison type is selected (see above), the Detune knob controls the phases of the signal duplicates instead of the pitch detuning.

← **In the "Phase Only" scenario it could also be a good idea to modulate the Detune parameter from an LFO using the Modulation Bus (see "The Modulation Bus section"), to create nice phasing effects.**

Spread

→ **Set the stereo spread amount of the signal duplicates.**

The User Wave and Mixer section



The User Wave and Mixer section is where you can load an external sample to use in the Oscillators (see "User Wave/User Wave Smooth") and/or in the Spectral Filter (see "User Wave"). In the Mixer you can mix and pan the signals from the three Sound Engines before sending them to the Filter, Amp and Multi FX sections.

Select Sample/Browse Sample buttons

→ **Load a sample by clicking the Browse sample button, or by using the Up/Down buttons to scroll and load a sample from the currently selected folder.**

- **It's possible to load stereo samples. However, the sample will be automatically converted to mono in Europa.**

! **The Europa patch does not include the actual sample - only a reference to it. Therefore, the sample has to be stored/located separately on disk on your computer. If Europa cannot find the sample at the location specified in the patch, it will look in the default Samples folder (where the factory samples were installed on installation). This makes it easier to exchange sample-based patches with other Europa users - just place any exchanged samples in the Samples folder and they will be automatically found.**

! **If the sample length is a multiple of 2048 samples ("Serum compatible"), no pitch detection is being made in Europa. Then, Europa automatically assumes that 2048 samples is one complete waveform cycle (period). If the sample is not an exact multiple of 2048 samples, a pitch detection is being performed by Europa to determine the pitch. Longer samples (with a stable pitch) will render better pitch detection results, so don't use very short samples.**

Level

→ **Set the volume of the corresponding Sound Engine signal with the Level slider.**
Range: -Inf to +6.0dB.

Pan

→ **Set the panning of the corresponding Sound Engine signal in the stereo panorama.**

The Filter section



Routing buttons

→ **Click the red LED buttons to route the corresponding Sound Engine signals to the Filter section.**
If deactivated, the signal bypasses the Filter and goes straight to the Amp section, see "The Amplifier section".

Drive

→ **Turn the Drive knob to amplify and introduce an overdrive type of distortion to the Sound Engine signal(s) in the filter.**

Filter Type selector

→ **Click the Filter Type name box to select one of the following filter types:**

- **SVF HP 12dB**
 A state variable (SVF) highpass filter with a 12dB/octave slope. This filter is similar to the State Variable Filter in the Thor synthesizer.
- **SVF BP 12dB**
 A state variable (SVF) bandpass filter with 12dB/octave slopes. This filter is similar to the State Variable Filter in the Thor synthesizer.
- **SVF LP 12dB**
 A state variable (SVF) lowpass filter with a 12dB/octave slope. This filter is similar to the State Variable Filter in the Thor synthesizer.
- **SVF Notch**
 A state variable (SVF) notch filter. This filter is similar to the State Variable Filter in the Thor synthesizer.

- **Ladder LP 24dB**
 A ladder-type lowpass filter with a 24dB/octave slope. The resonance peak more narrow in this filter type than in the MFB LP 24dB filter (see below). The filter can be driven to self-oscillate.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **MFB LP 12dB**
 A multiple feedback (MFB) lowpass filter with a 12dB/octave slope.
- **MFB LP 24dB**
 A multiple feedback (MFB) lowpass filter with a 24dB/octave slope. The resonance peak is wider in this filter type that in the Ladder filter (see above). The filter can be driven to self-oscillate.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **MFB HP 24dB**
 A multiple feedback (MFB) highpass filter with a 24dB/octave slope.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **K35 LP 12dB**
 An "early MS-20 type" of lowpass filter with a 12dB/octave slope. The filter can be driven to self-oscillate.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

Reso

→ **Set the resonance amount.**
In the SVF Notch filter, the Reso knob controls the width of the notch - from wide to narrow.

Freq

→ **Set the cutoff frequency (for the HP and LP filter types) or the center frequency (for the BP filter type).**

Frequency Modulation

→ **Turn the Kbd knob to set the keyboard tracking amount.**
At 0% the filter is static and doesn't track the keyboard at all. At 100% the filter tracks the keyboard 1 semitone per note.

→ **Click the Frequency Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

⤑ **Use one of the Envelopes (see "The Envelopes section") as modulation source to create a Filter envelope.**

→ **Set the modulation amount with the Frequency Modulation Amount knob.**

→ **Turn the Velo knob to control the Frequency Modulation Amount from Keyboard Velocity.**

⤑ **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**

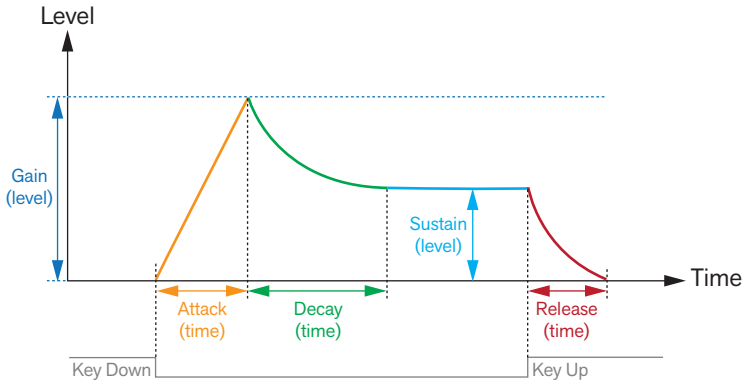
The Amplifier section



The Amplifier section contains a standard ADSR envelope, which controls the amplitude of the signals from all three Sound Engines equally.

- **To create an "amp envelope" for a separate Sound Engine, have a look at "Creating an individual "pre amp envelope" for a Sound Engine".**

The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

A(ttack)

When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Gain knob (see below). How long this should take, depends on the Attack setting. If the Attack is set to "0", the Gain level is reached instantly. If the Attack value is raised, it will take longer time before the Gain level is reached.

D(ecay)

After the Gain level has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.

S(ustain)

The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Gain level, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Gain level. Note that Sustain represents a level, whereas the other envelope parameters represent times.

R(elease)

The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

Pan

- **Set the panning of the output signal from the Amplifier in the stereo panorama.**
- **Since Pan works individually per voice, you can assign e.g. Keyboard Velocity or an Envelope in the Modulation Bus to control the Pan effect, see "The Modulation Bus section".**

Gain

- **Set the desired maximum level for the Amplifier with the Gain knob.**
 This is the maximum level the envelope will reach after the Attack stage is completed (see above).
- **If you want to create a tremolo effect, assign "Gain" as Destination and an LFO as Source in the Modulation Bus section, see "The Modulation Bus section".**

Velo

- **If you want the Gain level to be controlled from keyboard velocity, turn up the Velo knob.**

The Envelopes section



The Envelopes section features four separate polyphonic (one per voice) general purpose envelope generators, that can be assigned to control selectable parameter(s) in the Modulation Bus section.

The Envelopes are extremely flexible, and you can draw your own custom modulation shapes by clicking and drawing in the display area. There are also a number of preset shapes that you can use as starting points (or use as is). If you use Loop mode, you could turn the envelope into a kind of LFO.

See "The Modulation Bus section" for details on how to assign the Envelopes to the desired destination(s).

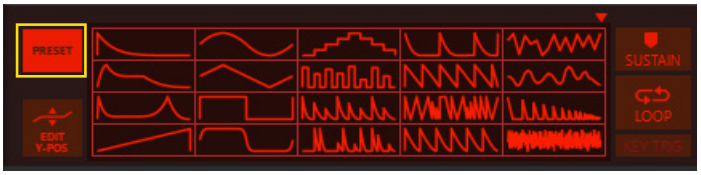
Envelope 1, 2, 3 and 4

→ **Click one of the Envelope 1, 2, 3 or 4 buttons to select which envelope to edit:**

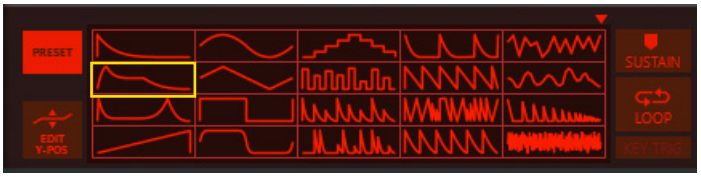


Preset

1. **Click the Preset button to bring up a palette of envelope preset curves:**



2. **Click the desired envelope preset curve to place it on the display.**
 Let's select a standard ADSR style of envelope curve:





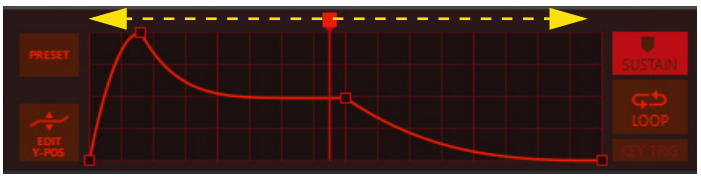
Adding a Sustain stage

→ **Click the Sustain button to add a sustain stage to the envelope:**



The vertical red marker that appears indicates at what level (and where) the envelope will stay sustained until you release the key.

→ **Drag the sustain marker sideways to move the sustain stage to the desired position:**



→ **To remove the sustain stage, click the Sustain button.**

Adding and removing envelope points

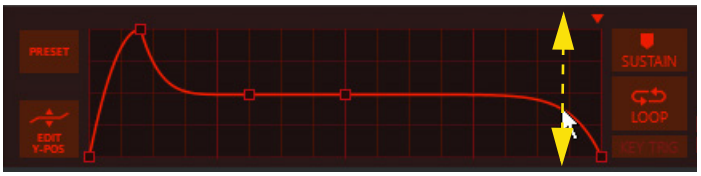
→ **Double click, or hold down [Ctrl](win) or [Cmd](Mac) and click in the envelope display to add points to the envelope curve:**



→ **To remove a point, double click, or hold down [Ctrl](win) or [Cmd](Mac) and click, on an existing point on the envelope curve.**

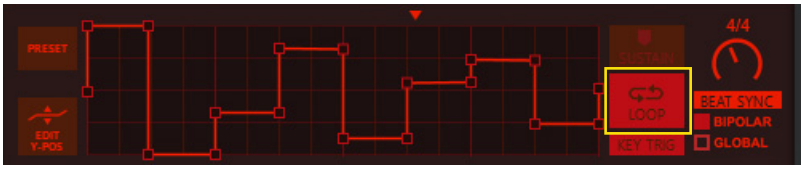
Changing the envelope curve shape

→ **Click a line segment (between two points) and drag up/down to change the curve shape:**



Looping the envelope

→ **Click the Loop button to turn the envelope into a kind of LFO.**
If there was previously a sustain stage in the envelope, this will automatically be disabled when you click the Loop button.

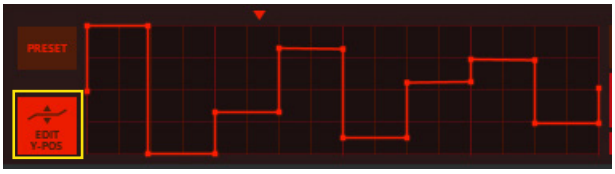


Here we have edited a stepped curve from the Presets. We have also enabled Beat Sync and set the length/rate to 4/4. This means that each step in the curve now represents an 1/8th note.

- **Key Trig means the envelope restarts when you play a note.**
- **You can choose whether the envelope should send out a bipolar value or unipolar one (0-100%).**
- **If Global is on, the envelope will be common for all voices.**

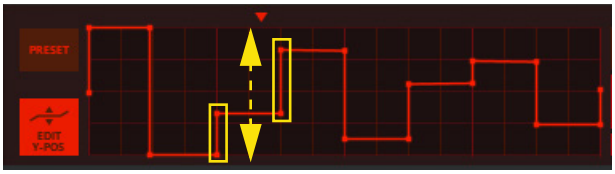
Editing levels only

→ **To restrict the editing to levels only, without affecting the time positions, click the Edit Y-Pos button:**



In this mode you cannot change the time positions of the envelope points, only their levels (height). This is extra useful with a stepped Preset curve, because dragging up or down will change the value of an entire segment, turning the Envelope into a pseudo-sequencer.

! **To be able to adjust the level of a segment, the two points on either side of the segment have to be on the exact same time positions. Otherwise, only the closest point will be changed. Also, any inclining/declining segment will automatically turn horizontal when edited:**

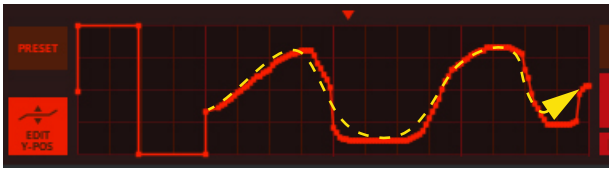


Adjusting the level of a segment.

Creating “free form” envelope curves

In the Edit Y-Pos mode, you can also draw “free form” curves:

→ **To continuously add new consecutive points, hold down [Ctrl](win) or [Cmd](Mac) and drag in the envelope display:**



→ **To erase points, hold down [Shift] and [Ctrl](win) or [Cmd](Mac) and drag in the envelope display.**

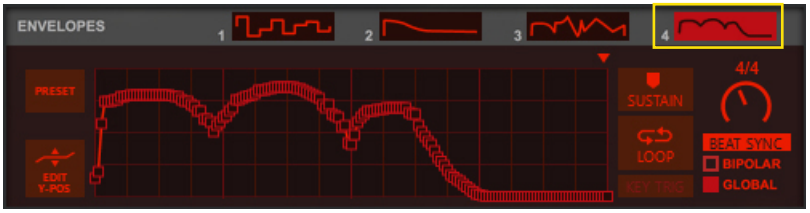
Using the Envelope 3 and Envelope 4 curves as Sound Engine waveforms

As a special feature you can use the Envelope 3 and Envelope 4 curves as waveforms for the Sound Engines:

1. **Select Envelope 3 and create/modify a curve in the envelope display:**



2. **Select Envelope 4 and create/modify another curve:**



3. **In the Waveform selector for a Sound Engine, select the "Envelope 3-4" waveform:**



4. **Turn the Shape knob to crossfade between the curves/waveforms of Envelopes 3 and 4:**



Using the Envelope 4 curve as a Spectral Filter curve

Another special feature is that you could use the Envelope 4 curve as a filter curve in the Spectral Filter:

1. **Select Envelope 4 and create/modify a curve in the envelope display:**



2. **In the Filter selector in the Spectral Filter section, select "Envelope 4":**



3. **Turn the Freq knob to change the curve's "cutoff" frequency and the Reso knob to change the curve's "resonance".**
 At Reso=0% the curve is completely flat (no gain or attenuation) and at Reso=100% the resonance corresponds exactly to the Envelope 4 curve.

The LFO section



An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of a signal to produce vibrato, but there are countless other applications for LFOs.

The LFO section features three separate general purpose LFOs, that can be assigned to control selectable parameter(s) in the Modulation Bus section, see "The Modulation Bus section".

1. **Select which of the three LFOs you want to edit by clicking one of the LFO 1, LFO 2 and LFO 3 buttons.**
2. **Select an LFO waveform by clicking the spin controls to the right of the waveform display, or by click-holding in the display and dragging up or down.**
 Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.
3. **Set the LFO frequency with the Rate knob.**

- **Click the Beat Sync button to sync the LFO to the main sequencer Tempo in Reason.**
 The Rate parameter now controls the time divisions.
- **Click the Key Sync button to restart the LFO at every new Note On.**
- **Click the Global button to make the LFO common for all voices (monophonic).**
- **Turn the Delay knob to introduce a delay before the LFO modulation kicks in after a note is played.**
 Turn clockwise for longer delay times.

The Effects section



The Effects section features six different effect modules that can be freely reordered by dragging & dropping. Most of the effect parameters are also available as destinations in the Modulation Bus, see "The Modulation Bus section".

At the top of the Effects section are six Effect buttons. Click any of these to bring up the control panel for the corresponding effect. Below the Effect buttons are the On/Off buttons for the individual effects. Click these to activate the effects.

Reordering the effects

→ **To define the order of the effects in the serial chain, click and hold on the desired Effect button and drag sideways to the desired position:**



Moving the Reverb effect to another position in the effects chain.

You can reorder the effects at any time.

Reverb



This is a stereo reverb, routed as a send effect.

- **Decay**
 This governs the length of the reverb effect.
- **Size**
 Sets the emulated room size, from small room to large hall. Middle position is the default room size.
 Lowering this parameter results in a closer and gradually more "canned" sound. Raising the parameter results in a more spacey sound, with longer pre-delay.
- **Damp**
 Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.
- **Amount**
 Use this parameter to adjust the send level to the Reverb effect.

 If you play a note, have a long delay Decay time and turn down Amount, the reverberation will continue.

Delay



This is a stereo delay, routed as a send effect.

- **Sync**
 Activate Sync to sync the delay time to the main sequencer Tempo.
- **Time**
 This sets the time between the delay repeats. If Sync is active (see above), the Time parameter now controls the time divisions.
- **Ping Pong**
 Activate Ping Pong to have the delay repeats alternating between left and right in the stereo panorama. The effect is also dependent on the Pan parameter (see below).
- **Pan**
 Sets the panning of the delay repeats in the stereo panorama. If Ping Pong is active (see above) the Pan knob controls the panning of the initial delay repeat as well as the total stereo spread of the remaining repeats.
- **FB**
 The FB (feedback) parameter determines the number of delay repeats.
- **Amount**
 Use this parameter to adjust the send level to the Delay effect.

► **If you play a note, have a long delay feedback and turn down Amount, the echoes will continue. This allows for automated "triggered delay" fx.**

Distortion



The Distortion effect features six different types of distortion.

→ **Select distortion type with the switch.**
"Dist" produces a dense, rich analog type of distortion.
"Scream" produces a less bright type of distortion.
"Tube" emulates a tube type of distortion.
"Sine" is a sine shaping distortion.
"S/H" gives the effect of sample rate reduction.
"Ring" is a ring modulator effect.

- **Drive**
 Sets the overdrive/feedback level of the selected distortion.
- **Tone**
 This is a lowpass filter and sets the tone of the selected distortion.
- **Amount**
 Sets the Dry/Wet amount of the distortion.

Compressor



This is a stereo compressor.

- **Attack**
 This governs how quickly the compressor will apply its effect when signals rise above the set threshold. If you raise this value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds.
- **Release**
 When the signal level drops below the set threshold, this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.
- **Thres**
 This is the threshold level above which the compression sets in. Signals with levels above the threshold will be affected, signals below it will not. In practice, this means that the lower the Threshold setting, the more the compression effect.
- **Ratio**
 This specifies the amount of gain reduction applied to the signals above the set threshold.

Phaser/Flanger/Chorus



This is a stereo Phaser/Flanger/Chorus.

→ **Select effect type with the Phaser/Flanger/Chorus switch.**
The selected effect type is displayed on the Effect button.

- **Depth**
 Sets the depth of the selected effect. To get a static sound, set Depth to zero.
- **Rate**
 Sets the rate/speed of the modulation.
- **Spread**
 Sets the stereo width of the effect.
- **Amount**
 Sets the Dry/Wet amount of the effect.

← **It's also possible to modulate the start/center frequency of the Phaser/Flanger/Chorus using the "Chorus/Flanger/Phaser > Frequency" destination in the Modulation Bus, see "The Modulation Bus section".**

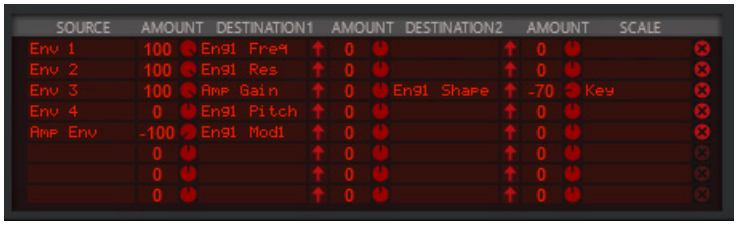
EQ



The EQ effect is a single band parametric equalizer with adjustable Q-value and Gain.

- **Freq**
 Sets the center frequency of the EQ band.
- **Q**
 Sets the bandwidth of the EQ band, from wide to narrow.
- **Gain**
 Sets the gain/attenuation of the EQ band, from -18dB to +18dB.

The Modulation Bus section



The Modulation Bus section is used for routing a modulation Source to one or two modulation Destinations each. This creates a very flexible routing system that complements the “pre-wired” routing in Europa.

The Modulation Bus section in Europa is derived from the one in the Reason Thor Polysonic Synthesizer device, so if you are familiar with Thor, you will quickly find your way around in Europa’s modulation bus.

There are eight “Source –> Destination 1 –> Destination 2 –> Scale” busses, of which the first four have pre-assigned sources. However, these four pre-assigned sources can be easily changed if you like.

A Source parameter can modulate two different Destination parameters per bus (with variable Amount settings). Each bus also has a Scale parameter that affects the relative modulation Amount for both Destinations.

► **Note that it is possible to assign the same source parameter as Source in several busses. This allows you to control more than two Destination parameters from the same Source.**

1. Select the desired Source parameter by clicking in the corresponding Source box and selecting from the list.

The following parameters can be used as modulation Sources:

Parameter	Description
Velocity	This applies modulation according to the Keyboard Velocity values (how hard or soft you strike the MIDI keyboard keys).
LFOs (LFO 1, LFO 2 and LFO 3)	This allows you to modulate parameters from LFO 1, LFO 2 and LFO 3 respectively.
Envelopes (Amp Envelope, Envelope 1, Envelope 2, Envelope 3, Envelope 4, Envelope 3 * Envelope 4, Envelope 3 * LFO 3)	This allows you to modulate parameters from any of the Envelopes. As a special feature you can also modulate parameters from the multiplied signal of Envelope 3 and Envelope 4, as well as from the multiplied signal of Envelope 3 and LFO 3.
Mod Wheel	This allows you to modulate parameters from the Mod Wheel.
MW Latched	This allows you to modulate parameters based on the current Mod Wheel value at a given Note On.
Pitch Wheel	This allows you to modulate parameters from the Pitch Bend control.

Parameter	Description
Breath	This allows you to modulate parameters from the Breath performance controller
Expression	This allows you to modulate parameters from the Expression performance controller
Aftertouch	This allows you to modulate parameters from Keyboard Aftertouch (channel aftertouch)
Sustain	This allows you to modulate parameters from a connected sustain pedal. Note that continuous sustain data (0-127) is supported - not just on/off.
Key	This is keyboard tracking. If a positive Amount value is used and the destination is filter frequency, the filter frequency will track the keyboard, i.e. increase with higher notes.
Random	This sends out a random value each time a new note is played.
Key In Octave	This allows you to modulate parameters based on 12 separate note values (within each octave).
Noise	This allows you to modulate parameters from white noise.
Polyphony	This allows you to modulate parameters based on the number of playing voices at a given time.
Last Velocity	This applies modulation according to the latest Keyboard Velocity value (how hard or soft you hit the latest MIDI keyboard key).
CV Input 1/2/3/4	This takes the current value on the CV 1/CV 2/CV 3/CV 4 inputs on the rear panel and sends to the desired destination. (Only available in the native Reason version!)
CV Input 1/2/3/4 Latched	This allows you to modulate parameters based on the current CV 1/CV 2/CV 3/CV 4 value at a given Note On. (Only available in the native Reason version!)

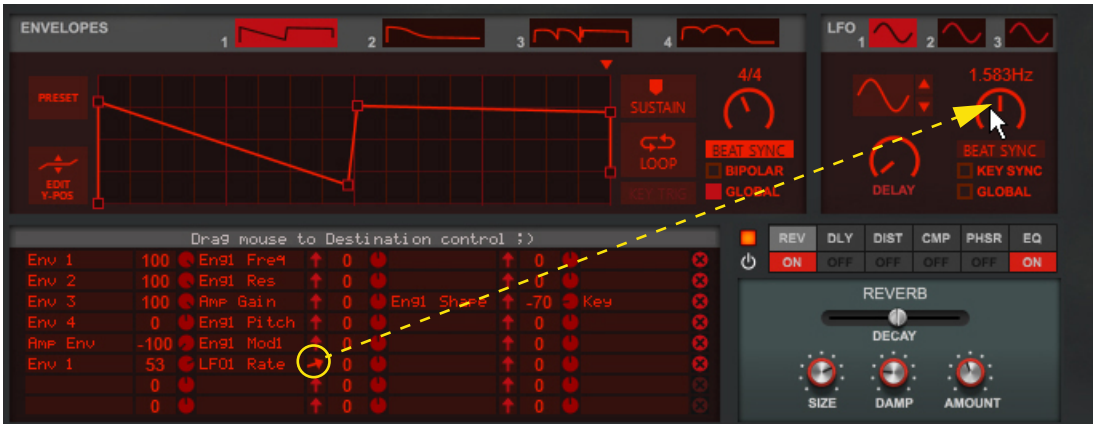
Modulation Bus Source parameters.

2. **Set the Amount for the first Destination by turning the corresponding Amount knob, or by clicking and dragging vertically in the corresponding Amount box.**

! **Note that the Amount range is +/-100. This means that the Amount value can exceed the modulated parameter's range. When this happens, the modulated parameter simply stays at its extreme value until the Amount value gets within the parameter's range again.**

3. **Select the first Destination parameter by click-holding the red arrow symbol to the right of the corresponding Destination box.**

4. **While click-holding, drag to the desired destination parameter on the panel:**



Assigning LFO 1 Rate as a Destination for Envelope 1.

As you hover over a valid destination control on the panel, the parameter name is automatically displayed in the Destination box in the Modulation Bus.

5. **To assign the currently selected Destination control, release the mouse button.**

→ **Alternatively, click the desired Destination box and select the Destination parameter from the list.**

The following parameters can be used as modulation Destinations:

Parameter	Description
Engine: Pitch	This affects the (full range) pitch of the Oscillator.
Engine: Shape	This affects the Shape parameter in the Oscillator section.
Engine: Mod 1 Amount	This affects the Modifier 1 Amount parameter in the Sound Engine.
Engine: Mod 2 Amount	This affects the Modifier 2 Amount parameter in the Sound Engine.
Engine: Filter Freq	This affects the Spectral Filter Frequency parameter in the Sound Engine.
Engine: Filter Res	This affects the Spectral Filter Resonance parameter in the Sound Engine.
Engine: Harmonics Pos	This affects the Harmonics Pos parameter in the Sound Engine.
Engine: Harmonics Amount	This affects the Harmonics Amount parameter in the Sound Engine.
Engine: Unison Count	This affects the Unison Count parameter in the Sound Engine.
Engine: Unison Detune	This affects the Unison Detune parameter in the Sound Engine.
Engine: Unison Blend	This affects the Unison Blend parameter in the Sound Engine.
Engine: Unison Spread	This affects the Unison Spread parameter in the Sound Engine.
Mixer: Level	This affects the Sound Engine Level in the Mixer section.
Mixer: Pan	This affects the Sound Engine Pan in the Mixer section
Filter: Drive	This affects the Drive parameter in the Filter section.
Filter: Freq	This affects the Frequency parameter in the Filter section.
Filter: Reso	This affects the Resonance parameter in the Filter section.
Amplifier: Gain	This affects the Gain parameter of the Amplifier section.
Amplifier: Pan	This affects the Pan parameter of the Amplifier section.
Amp Envelope: Attack	This affects the Attack time of the Envelope in the Amplifier section.
Amp Envelope: Decay	This affects the Decay time of the Envelope in the Amplifier section.
Amp Envelope: Sustain	This affects the Sustain level of the Envelope in the Amplifier section.
Amp Envelope: Release	This affects the Release time of the Envelope in the Amplifier section.
LFOs: Delay	This affects the LFO Delay parameters.
LFOs: LFO Rate	This affects the LFO Rate parameters.
Envelopes: Env Rate	This affects the Envelope Rate parameters.
Portamento	This affects the Portamento Time parameter.
CV Outputs: CV1/2/3/4 Out	This sends out the source modulation value(s) on the CV1/2/3/4 Output on the rear panel. (Only available in the native Reason version!)
Reverb: Decay	This affects the Decay parameter in the Reverb effect.
Reverb: Amount	This affects the Amount parameter in the Reverb effect.
Delay: Time	This affects the Time parameter in the Delay effect.
Delay: Feedback	This affects the FB parameter in the Delay effect.
Delay: Amount	This affects the Amount parameter in the Delay effect.
Delay: Pan	This affects the Pan parameter in the Delay effect.
Dist: Drive	This affects the Drive parameter in the Dist effect.
Dist: Tone	This affects the Tone parameter in the Dist effect.
Dist: Amount	This affects the Amount parameter in the Dist effect.
Compressor: Release	This affects the Release parameter in the Compressor effect.
Compressor: Ratio	This affects the Ratio parameter in the Compressor effect.
Chorus/Flanger/Phaser: Frequency	This affects the center frequency of the Chorus/Flanger/Phaser effects.
Chorus/Flanger/Phaser: Amount	This affects the Amount parameter of the Chorus/Flanger/Phaser effects.
Par EQ: Frequency	This affects the Freq parameter in the EQ effect.
Par EQ: Gain	This affects the Gain parameter in the EQ effect.

Modulation Bus Destination parameters.

6. **Set the Amount for the second Destination (if desired) by turning the corresponding Amount knob, or by clicking and dragging vertically in the Amount box for the second destination.**
7. **If desired, select a second Destination parameter by click-holding the blue arrow symbol to the right of the corresponding Destination box, and dragging to the desired control on the panel.**
8. **If desired, click the Scale box and select a Scale parameter.**
 The available Scale parameters are the same as the Source parameters, see "Modulation Bus Source parameters.".
9. **Turn the Scale Amount knob, or click the Amount box to the left of the Scale box and move the mouse pointer up or down to set a Scale Amount value.**
 Both positive and negative Scale Amount values can be set (+/- 100%). If you, for example, are using the Mod Wheel as Scale parameter and don't want any modulation when the Mod Wheel is set to zero, set the Scale Amount parameter to 100%. Then, there will be no effect when the Mod wheel is set to zero, and full modulation when the Mod Wheel is all the way up.

- **How much modulation will be applied when the Scale parameter is set to maximum is governed by the to Destination Amount parameter(s).**
- **How much the Scale parameter controls the modulation is set with the Scale Amount parameter.**

→ **To clear an assigned Source, Destination or Scale parameter, hold down [Ctrl](Win) or [Cmd](Mac) and click the Source/Destination/Scale box. Alternatively, click the Source/Destination/Scale box and select "Off" from the list.**

→ **To reset an Amount value to 0, hold down [Ctrl](Win) or [Cmd](Mac) and click the desired Amount box or knob.**

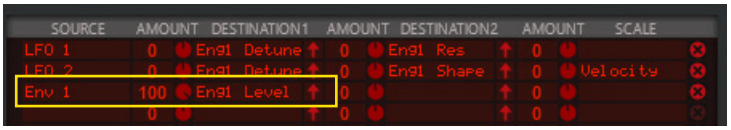
→ **To clear an entire modulation assignment (a whole row), click the circular X button to the right of the corresponding Scale box.**

Tips and Tricks

Creating an individual “pre amp envelope” for a Sound Engine

There might be situations where you want to control the amplitude envelopes separately for each Sound Engine. Let’s say you have a plucked sound with a fairly long release in Sound Engine 2 and then you want to slowly fade in a pad sound from Sound Engine 1. Since the built-in Amp Envelope controls all three Sound Engines together, you could use the following “workaround”:

1. **Assign “Envelope 1” as Source and “Mixer: Eng1 Level” as Destination in the Modulation Bus.**
 Turn the Amount knob to 100 in the Modulation Bus.



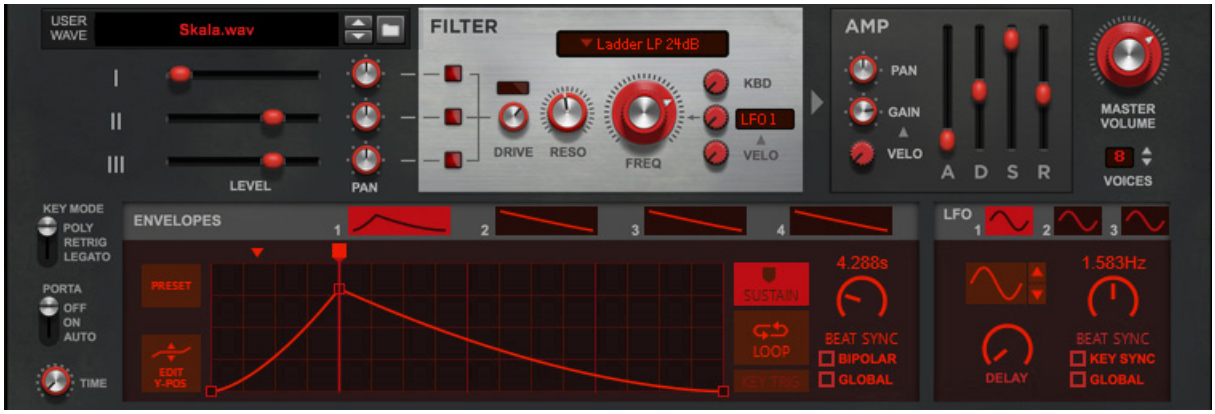
2. **Create your “fading” envelope curve in the Envelope 1 display.**
 Click the Sustain button in the Envelopes display if you want to have a sustain stage in your envelope.



3. **Set the Sound Engine 1 Mixer Level slider to zero.**



4. **As you play the keyboard, Envelope 1 will now fade in the signal from Sound Engine 1, while the signal from Sound Engine 2 is only controlled by the built-in Amp Envelope.**



! **Note that the built-in Amp Envelope’s settings will also affect the “fading pad” sound from Sound Engine 1, since all Sound Engine signals eventually pass through the Amp Envelope.**